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BORSETTI, GREG	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No. 10/582,025	Applicant(s) VIRETTE ET AL.	
	Examiner GREG A. BORSETTI	Art Unit 4141	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *08 June 2006*.
2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.
3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-26* is/are pending in the application.
4a) Of the above claim(s) _____ is/are withdrawn from consideration.
5)☐ Claim(s) _____ is/are allowed.
6)☒ Claim(s) *1-26* is/are rejected.
7)☐ Claim(s) _____ is/are objected to.
8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☒ The specification is objected to by the Examiner.
10)☒ The drawing(s) filed on *08 June 2006* is/are: a)☒ accepted or b)☐ objected to by the Examiner.
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
a)☒ All b)☐ Some * c)☐ None of:
1.☒ Certified copies of the priority documents have been received.
2.☐ Certified copies of the priority documents have been received in Application No. _____.
3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).
* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

1. This action is in response to preliminary amendment filed on 6/8/2006.

2. Claims 1-26 are pending.

Drawings

3. The drawings filed on 6/8/206 are accepted by the examiner.

Specification

4. The abstract of the disclosure is objected to because “The sheet or sheets presenting the abstract may not include other parts of the application or other material.” The abstract makes reference to function blocks of the drawings. Correction is required. See MPEP § 608.01(b).

The lengthy specification has not been checked to the extent necessary to determine the presence of all possible minor errors. Applicant's cooperation is requested in correcting any errors of which applicant may become aware in the specification.

Claim Objections

5. Claim 2 objected to under 37 CFR 1.75(c), as being of improper dependent form for failing to further limit the subject matter of a previous claim. Applicant is required to cancel the claim(s), or amend the claim(s) to place the claim(s) in proper dependent form, or rewrite the claim(s) in independent form. Claim 1 cites the execution of common functions of functional units in a common calculation module for at least some

of the coders. Claim 2 cites that the calculation module comprises at least one function unit of one of the coders, which does not further limit parent claim one because ***at least some*** still describes ***at least one***.

Claim Rejections - 35 USC § 112

6. The following is a quotation of the second paragraph of 35 U.S.C. 112:

> The specification shall conclude with one or more claims particularly pointing out and distinctly claiming the subject matter which the applicant regards as his invention.

Claim 14 rejected under 35 U.S.C. 112, second paragraph, as failing to set forth the subject matter which applicant(s) regard as their invention. Claim 14 cites an "at least partial coding functional unit" where it is not fully understood what exactly is meant by an at least partial coding and it is not described in the specification. Clarification is needed.

Claim 26 rejected under 35 U.S.C. 112, second paragraph, as being incomplete for omitting essential elements, such omission amounting to a gap between the elements. See MPEP § 2172.01. Claim 26 claims a device and is dependent on claim 25, which is a system claim. There is a gap between the system and the device because the device would be operative within the system and would not be functional without it. Claim 26 must be reworded to be functional as a system claim that further includes the claimed subject matter.

Claim Rejections - 35 USC § 101

7. 35 U.S.C. 101 reads as follows:

> Whoever invents or discovers any new and useful process, machine, manufacture, or composition of matter, or any new and useful improvement thereof, may obtain a patent therefor, subject to the conditions and requirements of this title.

Claim 24 of the claimed invention is directed to non-statutory subject matter. The claim specifically claims a software product, which is non-statutory under 35 U.S.C. 101. The examiner recommends amending the claim to read "A computer readable medium storing a computer program product in memory…" Correction is needed.

Claim Rejections - 35 USC § 103

8. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 1-12, 15-16, 21-22, and 24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kolesnik et al. (US Patent # 5729655 hereinafter Kolesnik) in view of Carter et al. (US Patent #5987506 hereinafter Carter)

As per claim 1, Kolesnik discloses:

- **identifying the functional units forming each coder and one or more functions implemented by each unit**
- [Kolesnik, column 12, lines 25-28] discloses "FIG. 4 shows an implementation of the variable rate LSP encoder 202. The LSP encoder 202 uses m quantized LSPs and comprises three schemes for LSP predicting and preliminary coding." Each of the preliminary coders are defined in [Kolesnik, columns 12-13, lines 25-67, 1-60] where the functional units are inherently known to the system.
- **marking functions that are common from one coder to another**
- [Kolesnik, column 13, lines 55-58] discloses "Generally, encoders 409 and 411 may use the same Huffman code, which differs from the code used by the encoder 1 407. The Huffman codes are precomputed using a large speech database." It would be obvious to someone of ordinary skill in the art that if the same Huffman codes are used, there would be operations that are functionally equivalent between encoders 409 and 411 such that the same code could be used because it is well known in the art that codewords are output from the Huffman coding where the data would have to be similarly comprised for the same function to apply.

Kolesnik fails to teach, but Carter teaches,

- **executing said common functions once and for all for at least some of the coders in a common calculation module**

- [Carter, column 18, lines 48-57] discloses "As further depicted in by FIG. 5, each node 212a-212c connects via the shared memory subsystem 220 to a virtual shared memory 222. As will be explained in greater detail hereinafter, by providing the shared memory subsystem 220 that allows the node 212a-212c to access the virtual shared memory 222, ***the computer network 210 enables network nodes 212a-212c to communicate and share functionality using the same techniques employed by applications when communicating between applications running on the same machine***." It is known in the art that distributed computing environments are capable of sharing functionality and information between operations in shared memory space. This is analogous to the instant application because the common calculation module would perform a singular function instead of reproducing it over and over throughout the coders. A distributed system would share the memory such that a single processor would process common information singularly, especially in the case of a dedicated processor.
- It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Carter with the Kolesnik device because "A further object of the invention is to provide computer network systems that have adaptable system configurations for dynamically exploiting distributed network resources and thereby increasing network performance and productivity [Carter, column 2, lines 58-62]. Carter provides a system that improves performance by reducing

redundancy which provides singularity in the Huffman code calculations of Kolesnik by sharing like information and processing.

As per claim 2, claim 1 is incorporated and Kolesnik does not specifically teach:

- **said calculation module comprises at least one functional unit of one of the coders**
- It would be inherent given the information in parent claim 1 that at least one function unit of the coders would operate within the calculation model because parent claim 1 states "executing said common functions once and for all for ***at least some*** of the coders in a common calculation module." ***At least some*** teaches ***at least one***.

As per claim 3, claim 2 is incorporated and Kolesnik teaches:

- **for efficient coding verifying an optimum criterion between complexity and coding quality**
- [Kolesnik, column 7, lines 49-51 discloses “To reduce the computational complexity of the search through the SCB, SCB analyzer 209 may be implemented as a trellis codebook…” Furthermore, [Kolesnik, column 5, lines 5-10] discloses “Compared to the Code Excited Linear Prediction (CELP) analyzer, one embodiment of the present invention reduces the number of bits needed for speech storing, or transmitting, without a significant loss in the

subjective speech quality." Kolesnik accounts for efficient coding to optimize the complexity and coding quality while reducing bit rate.

Kolesnik fails to teach, but Carter teaches,

- **for each function executed in step c), at least one functional unit is used of a coder selected from said plurality of coders and the functional unit of said coder selected is adapted to deliver partial results to the other coders**
- [Carter, column 18, lines 48-57] discloses "As further depicted in by FIG. 5, each node 212a-212c connects via the shared memory subsystem 220 to a virtual shared memory 222. As will be explained in greater detail hereinafter, by providing the shared memory subsystem 220 that allows the node 212a-212c to access the virtual shared memory 222, ***<u>the computer network 210 enables network nodes 212a-212c to communicate and share functionality using the same techniques employed by applications when communicating between applications running on the same machine</u>***." The information is stored in shared memory and is available to all processes needing to access it. Thus, Carter provides the sharing of results between the coders of Kolesnik.
- It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Carter with the Kolesnik device because "A further object of the invention is to provide computer network systems that have adaptable system configurations for dynamically exploiting distributed network resources and thereby increasing network performance and productivity [Carter, column 2, lines 58-62]. Carter provides a system that improves performance by reducing

redundancy which provides singularity in the Huffman code calculations of Kolesnik by sharing like information and processing.

As per claim 4 and 5, claim 3 is incorporated and Kolesnik teaches:

- **the selected coder is the coder with the lowest (highest) bit rate and the results obtained after execution of the function in step c) with parameters specific to the selected coder are adapted to the bit rates of at least some of the other coders by a focused parameter search for at least some of the other modes up to the coder with the highest (lowest) bit rate**
- [Kolesnik, column 8, lines 18-25] discloses "Since different excitation search modes require differing numbers of bits for excitation coding, the bit rate value is variable from frame to frame. The largest number of bits is required by SACBS mode while the smallest ACB mode is required. To reduce, or to limit, the bit rate, without a substantial loss in speech quality, some restrictions on the search mode usage may be imposed optionally." Then, [Kolesnik, column 8, lines 45-62] describes search mode selection involving "weighting coefficients effect the probability that a certain mode will be chosen for a given subframe. Through empirical study, the weighting coefficient of Table 2 have been found to provide subjectively good quality speech with a minimum average data rate…." Kolesnik provides bit rate adjustment using the weighting coefficients which, in effect, provides an equivalent step in varying the bit rate based upon the searching mode that is chosen for the coder. It would be obvious given the

information in claim 3 that the weighting coefficients would be shared across chosen coders such that coders of different bit rates would be accounted for whether the initial rate was highest or lowest.

As per claim 6, claim 4, is incorporated and Kolesnik teaches:

- **the functional unit of a coder operating at a given bit rate is used as the calculation module for that bit rate and at least some of the parameters specific to that coder are progressively adapted:**
  - **up to the coder with the highest bit rate by focused searching**
  - **up to the coder with the lowest bit rate by focused searching**
- [Kolesnik, column 8, lines 18-25] discloses “Since different excitation search modes require differing numbers of bits for excitation coding, the bit rate value is variable from frame to frame. The largest number of bits is required by SACBS mode while the smallest ACB mode is required. To reduce, or to limit, the bit rate, without a substantial loss in speech quality, some restrictions on the search mode usage may be imposed optionally.” Then, [Kolesnik, column 8, lines 45-62] describes search mode selection involving "weighting coefficients effect the probability that a certain mode will be chosen for a given subframe. Through empirical study, the weighting coefficient of Table 2 have been found to provide subjectively good quality speech with a minimum average data rate….” Kolesnik provides bit rate adjustment using the weighting coefficients which, in effect, provides an equivalent step in varying the bit rate based upon

the searching mode that is chosen for the coder. It would be obvious given the information in claim 3 that the weighting coefficients would be shared across chosen coders such that coders of different bit rates would be accounted for whether the initial rate was highest or lowest.

As per claim 7, claim 1 is incorporated and Kolesnik teaches:

- **the functional units of the various coders are arranged in a trellis with a plurality of possible paths in the trellis, wherein each path in the trellis is defined by a combination of operating modes of the functional units and each functional unit feeds a plurality of possible variants of the next functional unit**
- [Kolesnik, column 14, lines 18-24] discloses "The block diagram in FIG. 5 shows an implementation of a multi-mode trellis encoding and linear prediction (MM-CELP) speech synthesizer. The synthesizer accepts compressed speech data as input and produces a synthesized speech signal. The structure of the synthesizer corresponds to that of the analyzer of FIG. 2, except that trellis encoding has been used." Kolesnik discloses the use of a trellis coding structure in which the analyzer of Fig. 2 also uses the trellis structure. The analyzer of Fig. 2 provides variable rate LSP encoder 202 (Fig. 4). Kolesnik thus teaches the use of a trellis structure for the coders where the trellis provides an interconnected structure connecting the various function units.

As per claim 8, claim 7 is incorporated and Kolesnik teaches:

- **a partial selection module is provided after each coding step conducted by one or more functional units capable of selecting the results supplied by one or more of those functional units for subsequent coding steps**
- [Kolesnik, column 12, lines 25-28] discloses “FIG. 4 shows an implementation of the variable rate LSP encoder 202. The LSP encoder 202 uses m quantized LSPs and comprises three schemes for LSP predicting and preliminary coding.” As shown in Fig. 4, there is a codeword selector (412) that teaches a partial selection module because it selects the results supplied by the function units of the variable rate encoders prior to the encoding (213) as shown in Fig. 2A where Fig. 2A highlights the variable rate LSP encoder (202) in general.

As per claims 9 and 10, claim 7 is incorporated and Kolesnik teaches;

- **for a given functional unit, the path selected in the trellis is that passing through the lowest bit rate functional unit and the results obtained from said lowest (highest) bit rate functional unit are adapted to the bit rates of at least some of the other functional units by a focused parameter search for at least some of the other functional units up to the highest (lowest) bit rate functional unit**
- [Kolesnik, column 14, lines 18-24] as shown in claim 7 describes how a trellis structure is applied to Kolesnik in accordance with the instant application. Furthermore, it has been shown in claim 5 ([Kolesnik, column 8, lines 18-25]

and [Kolesnik, column 8, lines 45-62]) that Kolesnik provides bit rate adjustment using the weighting coefficients which, in effect, provides an equivalent step in varying the bit rate based upon the searching mode that is chosen for the coder. It would be obvious given the information in claim 3 that the weighting coefficients would be shared across chosen coders such that coders of different bit rates would be accounted for whether the initial rate was highest or lowest.

As per claim 11, claim 9 is incorporated and Kolesnik teaches:

- **the functional unit operating at said given bit rate is used as the calculation module and at least some of the parameters specific to that functional unit are progressively adapted:**
    - **up to the functional unit capable of operating at the lowest bit rate by focused searching**
    - **up to the functional unit capable of operating at the highest bit rate by focused searching**
- [Kolesnik, column 8, lines 18-25] discloses "Since different excitation search modes require differing numbers of bits for excitation coding, the bit rate value is variable from frame to frame. The largest number of bits is required by SACBS mode while the smallest ACB mode is required. To reduce, or to limit, the bit rate, without a substantial loss in speech quality, some restrictions on the search mode usage may be imposed optionally." Then, [Kolesnik, column 8, lines 45-62] describes search mode selection involving "weighting coefficients

effect the probability that a certain mode will be chosen for a given subframe. Through empirical study, the weighting coefficient of Table 2 have been found to provide subjectively good quality speech with a minimum average data rate…." Kolesnik provides bit rate adjustment using the weighting coefficients which, in effect, provides an equivalent step in varying the bit rate based upon the searching mode that is chosen for the coder. It would be obvious given the information in claim 3 that the weighting coefficients would be shared across chosen coders such that coders of different bit rates would be accounted for whether the initial rate was highest or lowest.

As per claim 12, claim 1 is incorporated and Kolesnik fails to teach:

- **said calculation module is independent of said coders and is adapted to redistribute results obtained in step c) to all the coders**

However, Carter teaches the above limitation,

- [Carter, column 18, lines 48-57] discloses "As further depicted in by FIG. 5, each node 212a-212c connects via the shared memory subsystem 220 to a virtual shared memory 222. As will be explained in greater detail hereinafter, by providing the shared memory subsystem 220 that allows the node 212a-212c to access the virtual shared memory 222, ***the computer network 210 enables network nodes 212a-212c to communicate and share functionality using the same techniques employed by applications when communicating between applications running on the same machine***." It is known in the art

that distributed computing environments are capable of sharing functionality and information between operations in shared memory space. This is analogous to the instant application because the common calculation module would perform a singular function instead of reproducing it over and over throughout the coders. A distributed system would share the memory such that a single processor would process common information singularly, especially in the case of a dedicated processor, which would remain independent of the rest of the processors analogous to the coders.

- It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Carter with the Kolesnik device because “A further object of the invention is to provide computer network systems that have adaptable system configurations for dynamically exploiting distributed network resources and thereby increasing network performance and productivity [Carter, column 2, lines 58-62]. Carter provides a system that improves performance by reducing redundancy which provides singularity in the Huffman code calculations of Kolesnik by sharing like information and processing.

As per claim 15, claim 1 is incorporated and Kolesnik teaches:

- **the coders in parallel are adapted to operate multimode coding and an a posteriori selection module is provided capable of selecting one of the coders**

- [Kolesnik, Fig. 2A] shows a parallel multi-mode coding scheme and the comparator and controller (210) is shown to select the mode.

As per claim 16, claim 15 is incorporated and Kolesnik teaches:

- **a partial selection module is provided that is independent of the coders and able to select one or more coders after each coding step conducted by one or more functional units**
- [Kolesnik, column 5, lines 23-24] discloses "In one embodiment, a set of admissible modes is determined based upon the mode used in the previous subframe." The comparator and controller (210) is independent of the coders and able to select the mode of the coders after the coding step of the previous frame is complete which teaches the after each coding step conducted by one or more functional units in the instant application.

As per claim 21, claim 1 is incorporated and Kolesnik teaches:

- **the coders are of the analysis by synthesis type and the method includes steps common to all the coders including:**
  - **preprocessing**
  - [Kolesnik, column 5, lines 53-57] discloses "The digital speech signal, which is typically sampled at 8 KHz, is first processed by a digital pre-filter 200. The purpose of such pre-filtering, coupled with the corresponding post-filtering, is to diminish specific synthetic speech noise." The

preprocessing of filtering the synthetic speech noise is common to all the coders.

- **linear prediction coefficient analysis**
- [Kolesnik, column 5, lines 10-13] discloses "Compared to the Code Excited Linear Prediction (CELP) analyzer, one embodiment of the present invention reduces the number of bits needed for speech storing, or transmitting, without a significant loss in the subjective speech quality. These advantages are achieved by: using three different excitation search modes, instead of two modes employed in CELP, together with a special strategy of mode selection, and by using ***an efficient LPC coding***." The LPC coding would inherently include LPC analysis.
- **weighted input signal calculation**
- [Kolesnik, column 6, lines 41-43] discloses "As in CELP, perceptual weighting is realized by passing the prefiltered speech signals through the weighting filter (WF)." The input signals are weighted in a filter to reduce speech noise lying in audible regions.
- **quantization for at least some of the parameters**
- [Kolesnik, column 5, lines 60-64] discloses "Pre-filtered speech is analyzed by short-term prediction analyzer 201. Short-term prediction analyzer 201 includes a linear prediction analyzer, a converter from linear prediction coefficients (LPC) into line spectrum pairs (LSPs) and a

quantizer of the LSPs." The line spectrum pairs are parameters and are quantized.

As per claim 22, claim 21 is incorporated and Kolesnik teaches:

- **the partial selection module is used after a split vector quantization step for short-term parameters**
- [Kolesnik, column, lines] discloses "Pre-filtered speech is analyzed by short-term prediction analyzer 201. ***Short-term prediction analyzer 201 includes a linear prediction analyzer***, a converter from linear prediction coefficients (LPC) into line spectrum pairs (LSPs) and a quantizer of the LSPs." Kolesnik analyzes short-term parameters prior to the partial selection module as defined above. It would be obvious to someone of ordinary skill in the art that split vector quantization could be used to analyze the short-term parameters because it is well known in the art. This can be seen in [Kolesnik, column 3, lines 12-16] discloses " The most effective approaches of this type are split-vector quantization, disclosed in "Efficient Vector Quantization of LPC Parameters at 24 bits/frame," K. K. Paliwal and B. S. Atal, Proceedings of the 1991 IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 661-664, May 1991..."

Claim 24 is rejected under the same principles as claim 1 because claim 1 provides the method for which the software product operates. Additionally, [Kolesnik,

claim 24] cites "a method of encoding digitized voice signals in a computer system...." It would be obvious that a method implemented within a computer system would be executable code embodied in a computer program product stored on a memory.

Claims 13-14, 23, 25-26 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kolesnik et al. (US Patent # 5729655 hereinafter Kolesnik) in view of Carter et al. (US Patent #5987506 hereinafter Carter) and further in view of Jabri et al. (US Patent #6829579 hereinafter Jabri).

As per claim 13, claim 12 is incorporated and Kolesnik fails to teach:

- **the independent module and the functional unit or units of at least one of the coders are adapted to exchange results obtained in step c) with each other and the calculation module is adapted to effect adaptation transcoding between functional units of different coders**

Jabri, in analogous art, teaches the above limitation,

- [Jabri, abstract] discloses "A method for transcoding a CELP based compressed voice bitstream from source codec to destination codec. The method includes processing a source codec input CELP bitstream to unpack at least one or more CELP parameters from the input CELP bitstream and interpolating one or more of the plurality of unpacked CELP parameters from a source codec format to a destination codec format if a difference of one or more of a plurality of destination codec parameters including a frame size, a

subframe size, and/or sampling rate of the destination codec format and one or more of a plurality of source codec parameters including a frame size, a subframe size, or sampling rate of the source codec format exist." Jabri provides a transcoding method between code excited linear prediction (CELP) based compression schemes. It is well known in the art that multi-mode coders can use different coders for different output means. This application, when applied to a known method of distributed computing and dedicated processing in Carter, would use the transcoding of Jabri to convert similar functions between coders such that they do not need to be repeated and processing time and coding complexity is reduced.

- Jabri and Kolesnik are analogous art because both deal with coding and compression of audio signals. It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Jabri with the Kolesnik device because "More particularly, the invention provides a method and apparatus for converting CELP frames from one CELP based standard to another CELP based standard, and/or within a single standard but a different mode. [Jabri, column 2, lines 8-12]" Since, multi-mode coders are known to use multiple coders for optimization, it would be obvious to someone of ordinary skill in the art that they would need to be transcoded to a uniform state to which they could be compared for the purposes of choosing the superior coding method for the given input signal.

As per claim 14, claim 12 is incorporated and Kolesnik teaches:

- **the independent module includes an at least partial coding functional unit**
- It would be obvious to someone of ordinary skill that some coding functions between different CELP schemes could be shared. The independent module, which provides equivalent functionality to multiple units for the purposes of processing reduction, would obviously have some sort of coding functional unit to have any significant impact on the reduction in processing.

Kolesnik fails to teach,

- **an adaptation transcoding functional unit.**

Jabri, in analogous art, teaches the above limitation,

- [Jabri, abstract] discloses "A method for transcoding a CELP based compressed voice bitstream from source codec to destination codec. The method includes processing a source codec input CELP bitstream to unpack at least one or more CELP parameters from the input CELP bitstream and interpolating one or more of the plurality of unpacked CELP parameters from a source codec format to a destination codec format if a difference of one or more of a plurality of destination codec parameters including a frame size, a subframe size, and/or sampling rate of the destination codec format and one or more of a plurality of source codec parameters including a frame size, a subframe size, or sampling rate of the source codec format exist." Jabri provides a transcoding method between code excited linear prediction (CELP) based compression schemes. It is well known in the art that multi-mode coders

can use different coders for different output means. This application, when applied to a known method of distributed computing and dedicated processing in Carter, would use the transcoding of Jabri to convert similar functions between coders such that they do not need to be repeated and processing time and coding complexity is reduced.

- Jabri and Kolesnik are analogous art because both deal with coding and compression of audio signals. It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Jabri with the Kolesnik device because “More particularly, the invention provides a method and apparatus for converting CELP frames from one CELP based standard to another CELP based standard, and/or within a single standard but a different mode. [Jabri, column 2, lines 8-12]” Since, multi-mode coders are known to use multiple coders for optimization, it would be obvious to someone of ordinary skill in the art that they would need to be transcoded to a uniform state to which they could be compared for the purposes of choosing the superior coding method for the given input signal.

As per claim 23, claim 21 is incorporate and Kolesnik fails to teach:

- **the partial selection module is used after a shared open loop long-term parameter search step**

Jabri, in analogous art, teaches the above limitation,

- [Jabri, column, lines] discloses “An open-loop pitch lag is estimated in every other subframe (except for the 5.15 and 4.75 kbit/s modes for which it is done once per frame) based on the perceptually weighted speech signal.” It would be obvious that if the open-loop long term parameters are based on the perceptually weighted speech signal, that they would be performed prior to the partial selection module in Kolesnik because the weighting is done directly after pre-filtering.
- Jabri and Kolesnik are analogous art because both deal with coding and compression of audio signals. It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Jabri with the Kolesnik device because “More particularly, the invention provides a method and apparatus for converting CELP frames from one CELP based standard to another CELP based standard, and/or within a single standard but a different mode. [Jabri, column 2, lines 8-12]” Since, multi-mode coders are known to use multiple coders for optimization, it would be obvious to someone of ordinary skill in the art that they would need to be transcoded to a uniform state to which they could be compared for the purposes of choosing the superior coding method for the given input signal.

Claim 25 is rejected under the same principles as claim 13. Claim 13 provides a transcoding method and claim 13 inherently incorporates claim 1, which provides the preparatory steps. Claim 25 further defines a memory which stores a software product.

[Kolesnik, claim 24] cites “a method of encoding digitized voice signals in a computer system...." It would be obvious that a method implemented within a computer system would be executable code embodied in a computer program product stored on a memory.

Claim 26 is rejected under the same principles as claim 12 because claim 12 provides the method for which the device in claim 26 operates.

Claims 17-20 are rejected under 35 U.S.C. 103(a) as being unpatentable over Kolesnik et al. (US Patent # 5729655 hereinafter Kolesnik) in view of Carter et al. (US Patent #5987506 hereinafter Carter) and further in view of Jabri et al. (US Patent #6829579 hereinafter Jabri) and further in view of Aguilar et al. (US Patent #7272556 hereinafter Aguilar).

As per claim 17, claim 1 is incorporated and Kolesnik fails to teach:

- **the calculation module includes a bit assignment functional unit shared between all the coders, each bit assignment effected for one coder being followed by an adaptation to that coder, in particular as a function of its bit rate**
- [Jabri, column, lines] discloses " Subframe interpolation may be needed when subframes for different sandards represent different time durations in the signal domain, or when a different sampling rate is used.” It would be obvious to someone of ordinary skill in the art that if different bit rates are used for the

coders, there would need to be an indication of the bit rate of the coder that would be common to all coders such that transcoding is possible. The bit assignment would be shared between all the coders in accordance with the distributed computing system in Carter.

- Jabri and Kolesnik are analogous art because both deal with coding and compression of audio signals. It would be obvious to someone of ordinary skill in the art at the time of the invention to combine Jabri with the Kolesnik device because “More particularly, the invention provides a method and apparatus for converting CELP frames from one CELP based standard to another CELP based standard, and/or within a single standard but a different mode. [Jabri, column 2, lines 8-12]” Since, multi-mode coders are known to use multiple coders for optimization, it would be obvious to someone of ordinary skill in the art that they would need to be transcoded to a uniform state to which they could be compared for the purposes of choosing the superior coding method for the given input signal.

Neither Kolesnik nor Jabri teaches:

- **the coders are of the transform type**

Aguilar, in analogous art, teaches the above limitation,

- [Aguilar, column 4, lines 1-3] discloses “Yet another object of the present invention is to provide a transform codec with multiple stages of increasing complexity and bit-rates.” Aguilar provides a transform coder in a multimode system.

- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.

As per claim 18, claim 17 is incorporated and Kolesnik teaches:

- **the method further includes a quantization step the results whereof are supplied to all the coders**
- Kolesnik would inherently include a quantization step which is supplied to the coders because a speech signal would be input as an analog signal which would be quantized and passed along to the coders for coding. The multimode coding scheme would have each coder receive the quantized signal such that the optimal coding scheme would be chosen.

As per claim 19, claim 18 is incorporated and Kolesnik fails to teach:

- **it further includes steps common to all the coders including:**
  - **a time-frequency transform**
  - **detection of voicing in the input signal**
  - **detection of tonality**
  - **determination of a masking curve**

- **spectral envelope coding**

Aguilar, in analogous art, teaches the above limitations,

- **a time-frequency transform**
- [Aguilar, column 8, lines 8-13] discloses "in accordance with the present invention, the band splitter 5 can be implemented as a filter bank, ***an FFT transform*** or wavelet transform computing device, or any other device that can split a signal into several signals representing different frequency bands." An FFT transform is a time-frequency transform.
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.
- **detection of voicing in the input signal**
- [Aguilar, column 10, lines 57-65] discloses "In speech applications it is usually necessary to provide a measure of how voiced (i.e., how harmonic) the signal is at a given time, and a measure of its volume or its gain. In very low bit-rate applications in accordance with the present invention one can therefore only transmit a harmonic frequency, a voicing probability indicating the extent to which the spectrum is dominated by voice harmonics, a gain, and a set of parameters which correspond to the spectrum envelope of the signal." Aguilar

provides a measure of how voiced the signal at a given time is, which inherently means it would be detected.

- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.
- **detection of tonality**
- [Aguilar, column 13, lines 60-64] discloses "The ***refined pitch estimate*** obtained in block 70 and the SEEVOC flat-top spectrum envelope are used to create in block 80 of the analyzer a smooth estimate of the spectral envelope using in a preferred embodiment cubic spline interpolation between peaks." The pitch estimate would inherently be a detection of tonality because by estimating the pitch estimation would determine a pitch average which would be indicative of the tonality of the speech or audio input.
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.

- **determination of a masking curve**
- [Aguilar, column 19, lines 35-37] discloses “In a preferred embodiment of the present invention, the masking envelope is computed as an attenuated LPC spectrum of the signal in the frame. This selection gives good results, since the LPC envelope is known to provide a good model of the peaks of the spectrum if the order of the modeling LPC filter is sufficiently high.” The masking envelope teaches a masking curve for eliminating low side effects.
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.
- **spectral envelope coding**
- [Aguilar, column 10, lines 48-51] discloses “The next block in FIG. 3A shows that instead of transmitting the magnitudes of each sinusoid, one can only transmit information about the spectrum envelope of the signal.” By transmitting the spectral envelope, it would inherently be coded.
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders.

Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.

As per claim 20, claim 17 is incorporated and Kolesnik fails to teach:

- **the coders effect sub-band and the method further includes steps common to all the coders including:**
  - **application of a bank of analysis filters**
  - **determination of scaling factors**
  - **spectral transform calculation**
  - **determination of masking thresholds in accordance with a psycho-acoustic model**

Aguilar, in analogous art, teaches the above limitations,

- **application of a bank of analysis filters**
- [Aguilar, column 8, lines 8-13] discloses "in accordance with the present invention, the band splitter 5 can be implemented ***<u>as a filter bank</u>***, an FFT transform or wavelet transform computing device, or any other device that can split a signal into several signals representing different frequency bands."
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders.

Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.

- **determination of scaling factors**
- [Aguilar, column 10, lines 54-57] discloses "As known in the art, the spectrum envelope can be encoded using different parameters, such as ***LPC coefficients, reflection coefficients (RC), and others***." The coefficients are scaling factors.
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.
- **spectral transform calculation**
- [Aguilar, column 17, lines 7-11] discloses "In the following block 35, the magnitude and unwrapped phase envelopes are upsampled to 256 points using linear interpolation in a preferred embodiment. Alternatively, this could be done using the Discrete Cosine Transform (DCT) approach described in Section E.1."
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders.

Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.

- **determination of masking thresholds in accordance with a psycho-acoustic model**
- [Aguilar, column 19, lines 17-21] discloses “Block 240 computes a masking envelope that provides a dynamic thresholding of the signal spectrum to facilitate the peak picking operation in the following block 250, and to eliminate certain low-level peaks, which are not associated with the harmonic structure of the signal.” The harmonic structure teaches the psycho-acoustic model and thus the masking envelope creates thresholds in accordance with a psycho-acoustic model.
- Aguilar and Kolesnik are analogous art because both pertain to multimode coding for audio signals. It would be obvious to someone of ordinary skill in the art to combine Aguilar with the Kolesnik device because Aguilar is an analogous invention which uses transform coders instead of CELP coders. Thus, it would be obvious to switch the coders for either transform or CELP coders because they are functionally equivalent elements.

Conclusion

9. Refer to PTO-892, Notice of References Cited for a listing of analogous art.

10. Any inquiry concerning this communication or earlier communications from the examiner should be directed to GREG A. BORSETTI whose telephone number is

(571)270-3885. The examiner can normally be reached on Monday - Thursday (8am - 5pm Eastern Time).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chameli Das can be reached on 571-272-3696. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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Dated: 5/7/08